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DATE: Wednesday, January 18, 2006

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☐	L26	L24 and SIP	1
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☐	L24	L23 and @AD<20020924	15
☐	L23	L22 and l10	19
☐	L22	(location or address) near8 (timestamp or (time stamp))	3442
☐	L21	l18 and l10	1
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☐	L19	L18 and @AD<20020924	2
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☐	L17	(contact address) near8 (timestamp or (time stamp))	1
☐	L16	L15 and l10	0
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☐	L14	l3 and (timestamp or (time stamp))	4
☐	L13	L11 and @AD<20020924	4
☐	L12	L10 and l3	1
☐	L11	L10 and l1	10
☐	L10	(deregistration or deregistrate) or ((remove or removing) address)	1320
☐	L9	L8 and l3	8
☐	L8	SIP proxy	744
☐	L7	L6 and @AD<20020924	0
☐	L6	L4 and l1	4
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☐	L4	SIP proxy user agent	8
☐	L3	L2 and @AD<20020924	18
☐	L2	((session initiation protocol) or SIP) near4 address near8 (register or registration or registrar) near4 message	51
☐	L1	((session initiation protocol) or SIP) near4 address near8 (register or registration or registrar)	192

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L9: Entry 3 of 8 File: PGPB Nov 14, 2002

DOCUMENT-IDENTIFIER: US 20020169887 A1
TITLE: System and method for assisting in controlling real-time transport protocol flow through multiple networks via screening

Application Filing Date:
20010427

Summary of Invention Paragraph:
[0023] Certain simple gateways, such as, but not limited to, the Cisco AS5300, can forward SIP-based call requests to a SIP proxy server. Unfortunately, these gateways have low densities and frequently lack the sophistication of softswitches in setting up routing policies. These routers, therefore, cannot be interconnected to create networks without a softswitch controller.

Brief Description of Drawings Paragraph:
[0046] FIG. 12A is a flow chart that illustrates steps taken by a SIP proxy to analyze a SIP message.

Detail Description Paragraph:
[0059] SIP is a protocol that has a number of key mechanisms defined. A first SIP mechanism is called a "register" message. When sent to a SIP proxy server, this message indicates that the endpoint is capable of receiving a communication for a specific user. This "register" message binds the physical IP address to the user using the IP address. A second SIP mechanism is the "invite" message. This message is sent to another endpoint to request a communication session. The "invite" message is sent all the way to the endpoint of the receiver of the communication. The receiver of the "invite" will then respond with an OK message indicating that the communication is accepted. When there are more than a few endpoints, or when there are endpoints that need certain features, a SIP proxy server acts as a go-between. The SIP proxy server receives and forwards the "invite" messages that are received for its users that have previously sent a "register" message.

Detail Description Paragraph:
[0060] FIG. 2 provides a detailed illustration of interaction between two SIP agents via a SIP proxy. For example, if a user sends a "register" message 242 from a first SIP user agent 244, a SIP proxy server 246 acknowledges the registration. Then, if a second SIP user agent 248 sends an first "invite" message 252 for the user that transmitted the "register" message" 242, the first "invite" message 252 is received by the SIP proxy server 246. The SIP proxy server 246 then transfers a second "invite" message 254 to the first SIP user agent 244. If the first SIP user agent 244 is willing to accept communication from the second SIP user agent 248, the first SIP user agent 244 transmits a message of approval to the SIP proxy server 246 which is then transmitted to the second SIP user agent 248.

Detail Description Paragraph:
[0061] A third SIP mechanism is the "bye" message, which unilaterally sends a communication session, and frees all of the network resources in use. Either side of a communication can send a "bye" message at any time. One notion embodied in the SIP architecture is that the user has mobility wherein the user can send a

L9: Entry 6 of 8 File: PGPB Sep 12, 2002

DOCUMENT-IDENTIFIER: US 20020126701 A1
TITLE: System and methods for using an application layer control protocol transporting spatial location information pertaining to devices connected to wired and wireless internet protocol networks

Application Filing Date:
20011030

Detail Description Paragraph:
[0145] Once the use has the server address it can send the SIP REGISTER message containing the SL information in its body. The Registrar server that behaves like the Authentication Server (AUS) required in the SLO architecture will authenticate the initial message. If the contacted SIP server is not a Location Server (LS) the message has a specific header for discovering the Location Server. It will be indicated in the SIP header "Require" sent in the REGISTER message. Hence, for indicating that the incoming message contains Spatial Information and needs to be processed by a Location Server, the SIP REGISTER will have the following header: "Require: SLO-server". It indicates that the user is registering his location and needs a Spatial Location Server to manage this data. In case that the server contacted has no SL capabilities the user will receive back a response where the "Contact:" header includes the address of the new SIP server, which can handle that message. An example would be like this: "Contact: slo-server.nokia.com".

Detail Description Paragraph:
[0151] Since the SLO is a new service, it remains a possibility that the contacted SIP registrar does not support this feature. There are two possibilities, the first one is trying to register with the closest SIP Registrar and wait for its response, and the other solution would be to use the OPTIONS message for querying beforehand the Registrar's capabilities. In the former case, if the registrar can handle a SLO message the registration will succeed, otherwise the User Agent receives a 300 response with the address of a SLO enabled SIP Registrar. In the other case, the User Agent needs to negotiate this capability with the registrar. The client will send an OPTIONS message to the registrar for indicating that he needs a SLO based registrar. The registrar can send back a 200 OK response, which means that it can manage this type of registration. Otherwise, the registrar returns a 300 Multiple Choices response, which means that the requested capability can be accessed through the proxy given by the "Contact:" field.

Detail Description Paragraph:
[0211] The CSCF (with SIP Proxy/Registrar capabilities) will accept the user registration. Furthermore, the CSSF sends the User Information either to the HSS (Signalling Interface Cx) or the Presence Server if such an entity.exists, as shown. After this the user is registered and his situation is available for other services. If the Presence Server does not exist, then the HSS will behave like a similar Presence Server. At this point is a matter or re-using or not overloading existing entities.

L26: Entry 1 of 1 File: PGPB Jan 23, 2003

DOCUMENT-IDENTIFIER: US 20030018704 A1
TITLE: Network presence and location agent

Application Filing Date:
20010308

Brief Description of Drawings Paragraph:
[0015] FIG. 8 shows a message flow for user deregistration;

Brief Description of Drawings Paragraph:
[0016] FIG. 9 is a high level block diagram showing how a SIP NPL agent may be employed in connection with an IP network;

Brief Description of Drawings Paragraph:
[0017] FIG. 10 shows a message flow for user registration, for a SIP based embodiment; and

Brief Description of Drawings Paragraph:
[0018] FIG. 11 shows a message flow for retrieval of presence/location information in response to a request from an application, for a SIP based embodiment.

Detail Description Paragraph:
[0027] Of course, it is possible to use other protocols for communication between the wireless network 26 and the NPL agent 21. For instance, rather than communicating with the wireless network 261 through the SMSC 28, the NPL agent 21 may communicate directly with the wireless network 26 using XML over SS7 (as shown in FIG. 2). As another example, TCP/IP may be used to connect to some of the network nodes directly. Similarly, Session Initiation Protocol (SIP) may be used, as described further below.

Detail Description Paragraph:
[0036] The push agent 47 publishes presence or location information to applications 44 without requiring applications 44 to request the information. The push agent 47 relies on the wireless service provider's HLR/ Mobile Switching Center (MSC) to proactively push notification of the user's registration or deregistration via the wireless network 43. In one embodiment, the presence/location information is pushed to the push agent 47 via TCP/IP in the form of a Serving System Message according to J-STD-025 (TIA/ATIS Internet Standard, "Lawfully Authorized Electronic Surveillance", December 1997). Alternatively, XML over TCP/IP may be used. The push agent 47 is also capable of handling other types of messages which contain the user presence and location information from the wireless network via TCP/IP. The messages are decoded by the push agent 47, and the presence/location information is published to the applications using XML over HTTP and TCP/IP.

Detail Description Paragraph:
[0058] In the case of user deregistration, a J-STD-025 Serving System Message will be generated by the HLR 46 when the MSC issues a MobileStationInactive message for a mobile device. The presence/location information will be published to the application 44. The process, as shown in FIG. 8, is as follows:

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L9: Entry 1 of 8 File: PGPB Feb 5, 2004

DOCUMENT-IDENTIFIER: US 20040024902 A1
TITLE: Megaco protocol with user termination

Application Filing Date:
20020618

Detail Description Paragraph:
[0019] The media gateway controller MGC is the part of the gateway which commands the media gateway to connect and release the connections. In other words, the MGC performs the control plane functions and comprises the intelligence. The media gateway controller may, for example, be controlled by a so called soft switch or a SIP (Session Initiation Protocol) proxy via which the actual signaling is routed, part of the soft switch or the SIP proxy, or it may be a network node via which the signaling is routed. Depending on how the MGC is implemented, it may be involved in signaling and may co-operate with other signaling protocols, such as SIP or it can receive control information with some protocol, such as Parlay API or SOAP (Simple Object Access Protocol, defined by the World Wide Web Consortium W3C). The media gateway controller according to the invention is described in more detail below with the connection model and with FIGS. 2 to 5. One media gateway controller controls one or more media gateways.

Detail Description Paragraph:
[0045] The CPS 11 is responsible for control-plane management of the PMR communications. Its important role may require various functionalities which can be implemented in the following modules: "PMR server"--the application that handles the sessions for group memberships which are signaled with an appropriate session control protocol, such as SIP, established for the PMRoC communications, and manages the users profiles (call rights, group active membership, scanning settings, etc.); SIP Proxy/Location Server--providing user location and routing functionalities of SIP signaling; SIP Registrar--for user registration/authentication; and Media Gateway Controller--controlling the network entities involved in the IP layer data distribution according to the group & user specific information (membership, rights, scanning settings, etc.).

Detail Description Paragraph:
[0068] Referring to FIG. 4, once the UE knows the IP address of the U-CPF it sends a SIP registration message 4-1 to the U-CPF. The SIP registration message contains among other things user's SIP URL, port number in terminal for One-to One calls and scanning settings for One-to-One calls. It may contain also the mnemonic that the user wants to use for One-to-One calls. The foregoing parameters are parameters that may be sent to the UT. In response to message 4-1, the U-CPF sends message 4-2, which contains an authorization challenge. The UE responds by sending message 4-1', which is the previously sent SIP registration message 4-1, with authentication response.

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L9: Entry 2 of 8 File: PGPB Jun 26, 2003

DOCUMENT-IDENTIFIER: US 20030120813 A1
TITLE: Apparatus and method for optimizing message sizes of textual protocols used in multimedia communications

Abstract Paragraph:
An apparatus and method for generating compressed SIP messages from full sized SIP messages and vice versa in order to decrease call set up time in an IP based communication system. During registration of a device, the invention caches the device's static information in the core network in a "Registrar/Location Server." Subsequently, during call set up, the device transmits its dynamic information to the SIP Agent in a compressed SIP message over an air interface. The SIP Agent retrieves the static information (from the Registrar/Location Server) along with the dynamic information in the compressed SIP message to generate a full sized SIP message. The SIP Agent forwards the full sized SIP message to a SIP Proxy, which is then transmitted to the IP system. Likewise, when a full sized SIP message is received from the IP system, the message is forwarded to the SIP Agent to generate a compressed SIP message for ultimate transmission to the device over the air interface.

Application Filing Date:
20011221

Detail Description Paragraph:
[0022] As previously stated, the static and default dictionary information is used by the MS 102 (138) and the SIP Agent 108 (124) to compress call setup messages for transmission over the air interface and to expand compressed messages received in the MS 102 (138) or SIP Agent 108 (124). The static and default dictionaries generated using the information in the Register Message of FIG. 3 are shown in FIG. 5. The static dictionary contains: 1) the contents of the first line (ss1.wcom.com; SIP/2.0); 2) the contents of the Via line (SIP/2.0/UDP ss1.wcom.com:5060); 3) the contents of the From line (BigGuy<sip:1-314-555-1111@ss1.wcom.com>)- ; 4) the contents of the Content-Type line (application/SDP); 5) the contents of the Content-Length line (132); 6) the contents of the v line (0); 7) the contents of the o line (UserA 2890844426 2890844426 IN IP4 ss1.wcom.com); 8) the contents of the s line (Session SDP); 9) the contents of the c line (IN IP4 100.101.102.104); and 10) the contents of the t line (0 0). The default dictionary contains: 1) the first contact address in the Register message (BigGuy<sip:UserA@here.com>); 2) the first m line in the Register message less the port number (m=audio . . . RTP/AVP 0); and 3) the first a line in the Register message (a=rtpmap:0 PCMU/8000).

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L9: Entry 5 of 8 File: PGPB Sep 19, 2002

DOCUMENT-IDENTIFIER: US 20020131395 A1
TITLE: Session initiation protocol (SIP) user agent in a serving GPRS support node (SGSN)

Application Filing Date:
20011219

Detail Description Paragraph:
[0033] As will be explained in greater detail below, the SIP application server 216 is a SIP-based service platform, i.e., may be implemented as a SIP proxy/redirect/registrar server enhanced with service logic execution environment, APIs, web server, and media servers. The web server in the SIP AS 216 provides SIP session event triggered web services/applications, as well as HTTP event triggered SIP session service such as click-to-dial. A media server can be an E-mail server or Short Message Service Center, etc. The media servers may not necessarily reside in the same box as the SIP AS 216. A media server provides SIP session event triggered multimedia services, or vice versa, media triggered SIP session services.

Detail Description Paragraph:
[0034] More particularly, FIG. 4 is a diagram illustrating an exemplary SIP application server 216 in accordance with an embodiment of the present invention. The SIP Application Server 216 is a service centric SIP Proxy/Redirect/Registrar Server to provide voice/web/email combined multimedia services. In the embodiment illustrated, the SIP Application Server 216 includes a Call Server 1100, a Web Server 1102, a Media Server 1104, and an Execution Environment 1106. The Media Server 1104 may not necessarily be an integral part of a SIP Application Server. In certain embodiments, the SIP Application Server 216 is able to interface with external Media Servers (not shown) via IETF protocols whether it has internal Media Servers or not.

Detail Description Paragraph:
[0037] As a SIP Proxy/Redirect/Registrar Server, the SIP Application Server 216 supports the SIP protocol to handle SIP requests and responses. In addition, it may support various service/application-orient- ed extensions to the baseline SIP. One is SIP extension of SIMPLE (SIP for Instant Messaging and Presence Leveraging) in order to support Presence and Instant Messaging services, as will be explained in greater detail below.

Detail Description Paragraph:
[0038] The Call Server 1100 is a SIP protocol engine that processes SIP requests and responses. The Call Server 1100 may support service/application-oriented extensions to the baseline SIP. Among these are the SUBSCRIBE and NOTIFY messages as specified in SIMPLE, which are used to implement Presence and other services/applications needed to dynamically arm and report a trigger event. The Call Server 1100 inter-works closely with the Execution Environment 1106, for example, in the same SIP Application Server. In typical cases, the Call Server 1100 would hand the incoming SIP message to the Execution Environment 1106 for service/application execution decision. The Execution Environment 1106 would then

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L13: Entry 1 of 4 File: PGPB Oct 14, 2004

DOCUMENT-IDENTIFIER: US 20040203763 A1
TITLE: Method of registering and deregistering a user

Application Filing Date:
20020327

Summary of Invention Paragraph:
[0008] It has been appreciated by the inventor that all of these checks are not required for deregistration. Thus, the use of the same message for registration and deregistration is disadvantageous in that unnecessary processing is required. This unnecessarily slows down the process of deregistration and also unnecessarily uses up the HSS resources.

Detail Description Paragraph:
[0026] In step S1, the user equipment 10 sends a register request from the user equipment to the P-CSCF 16. The purpose of this request is to register the users SIP (session internet protocol) uniform resource identifier with a CSCF 22 in the home network. This request is routed to the P-CSCF 16 because it is the contact point with the IP multimedia subsystem for the user equipment. The register message may include the following information: private identity, public identity, home domain name and the requested expiration time of the registration; content length, destination domain for the request; the IP address of SIP session allocated; the IP address of the user and authorisation information.

Detail Description Paragraph:
[0037] Reference will now be made to FIG. 3 which shows the procedure for the deregistration of an already registered user. Steps T1 to T5 are similar to steps S1 to S5 of FIG. 2. However, there is no need to do the S-CSCF selection as this will have already been done. As the user is already registered, the next step after step T5 will be steps T6 and T7 which are the same as steps S18 and S19 of FIG. 2.

Detail Description Paragraph:
[0039] Reference will now be made to an embodiment of the present invention. In embodiments of the present invention, the register procedure is as outlined in relation to FIG. 2. However, a different procedure is provided for the deregistration procedure. In his regard, reference is made to FIG. 4.

Detail Description Paragraph:
[0040] FIG. 4 shows the signal flow during a deregistration procedure. In the first step 01, a register message is sent from the user equipment to the P-CSCF 16. This is the same as step S1 in FIG. 2. In step Q2, the P-CSCF sends the register message to the I-CSCF 20. Again, this is the same as step S2 in FIG. 2.

Detail Description Paragraph:
[0041] Reference is made to FIG. 5 which shows the method carried out by the I-CSCF 20. In step A1, the register message is received in step A2, the I-CSCF 20 checks the value of then expires header field which contains the requested expiry time in the SIP REGISTER message In particular in step A3, the I-CSCF checks if the value is 0. If it is, the next step is A4. The value 0 is taken to indicate that the

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L12: Entry 1 of 1 File: PGPB Aug 29, 2002

DOCUMENT-IDENTIFIER: US 20020120760 A1
TITLE: Communications protocol

Application Filing Date:
20010529

Summary of Invention Paragraph:
[0034] In any case, when proxy/redirector 306 receives the INVITE message, it communicates with a registrar/location server 308 to retrieve the location (transport address) corresponding to the SIP-URL used to indicate the callee. Typically, registration is performed by a terminal device upon startup utilizing a REGISTER message. When acting as a proxy, server 306 establishes the call by sending an INVITE to terminal device 304 and continues to act as a go-between for the endpoints during the session. When acting as a redirector, server 306 returns the address of terminal device 304 to terminal device 300, which then establishes the session directly with terminal device 304. It should be noted that, while illustrated as two different machines, often times registrar 308 and proxy/redirector 306 are implemented on the same machine. Also, through the use of the registration server, SIP provides for terminal mobility, in addition to personal mobility.

Detail Description Paragraph:
[0215] When the client sends this transaction to the server, the server will accept the transaction (using BLRemove.accept) but all that means is that the server accepts the transaction--this does not mean that the client can remove the entry from it's address book. The client may remove an address book entry only when it receives a BLRemove message from the server.

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